

**IN THE CLAIMS:**

This listing of claims will replace all prior versions, and listings, of claims in the application:

**LISTING OF CLAIMS:**

- 1        1.    (currently amended) A method for determining whether to accept a new call to  
2        be routed from a first location to a second location via a network path in an IP  
3        network, comprising the steps of:  
4            (a) obtaining, at the first location, information relevant to the quality of service  
5        of voice calls being transmitted from the first location to the second location ~~via the~~  
6        ~~IP network~~ via the network path;  
7            (b) calculating, based on said information, a parameter indicative of a  
8        congestion status of the network path from the first location to the second location;  
9        and  
10        (c) accepting the new call into the IP network at the first location in the case of  
11        said parameter not exceeding an upper threshold.
- 1        2.    (original) The method of claim 1 wherein said new call is accepted into the IP  
2        network at a reduced bandwidth in the case of said parameter exceeding a lower  
3        threshold.
- 1        3.    (original) The method of claim 1 where said new call is not accepted into the  
2        IP network in the case of said parameter exceeding the upper threshold.
- 1        4.    (currently amended) The method of claim 1 wherein the information obtained  
2        is a number of sent packets transmitted from said first location to said second  
3        location ~~in the IP network~~ via the network path, wherein the number of sent packets  
4        comprises a number of lost packets, a number of late packets and a number of  
5        received packets.
- 1        5.    (currently amended) The method of claim 1 wherein the information obtained  
2        is a delay of received packets transmitted from said first location to said second  
3        location ~~in the IP network~~ via the network path.

6. (currently amended) The method of claim 1 wherein the information obtained is a delay variation of received packets transmitted from said first location to said second location ~~in the IP network~~ via the network path.

7. (original) The method of claim 1 wherein the information is obtained on a periodic basis.

8. (original) The method of claim 1 wherein the information is obtained on an exception basis using an immediate report.

9. (original) The method of claim 1 wherein the parameter is identified as a packet lost ratio (PLR).

10. (original) The method of claim 9 wherein PLR is defined as

$$PLR = \frac{(\text{lost packets} + \text{late packets})}{(\text{received packets} + \text{lost packets} + \text{late packets})}$$

11. (original) The method of claim 2 wherein bandwidth is reduced for a newly accepted call by selecting a first encoder to encode the new voice call information in a bandwidth that is smaller than bandwidths of other calls accepted in the network that are encoded by a second encoder.

12. (previously presented) The method of claim 2 wherein the bandwidth of a newly accepted call is reduced by increasing the packet size for said newly accepted voice call, wherein the packet size is indicative of a size of a corresponding voice sample.

13. (original) The method of claim 2 wherein the bandwidth of a newly accepted call is reduced by activating the characteristic of silence suppression for said newly accepted voice call.

14. (currently amended) Apparatus comprising a first gateway for interfacing voice call data from a public switch telephone network to an internet protocol network, said first gateway comprising:

4 a first circuit for passing said voice call data of voice calls to the internet protocol  
5 network;

6 a second circuit for receiving quality-of-service information associated with voice  
7 calls currently being transmitted toward a second gateway via the first circuit; and

8 a third circuit for:

9 calculating, based on the received quality-of-service information, a  
10 parameter indicative of a congestion status of a network path ~~associated with the first~~  
11 ~~circuit~~ between the first gateway and the second gateway; and

12 determining, by comparing said parameter to at least one threshold,  
13 whether a new voice call is to be accepted into the internet protocol network via the  
14 first circuit.

1 15. (original) The apparatus of claim 14 wherein said first circuit further comprises  
2 one or more Ethernet cards that are connected to the internet protocol network.

1 16. (original) The apparatus of claim 14 wherein said second circuit is at least one  
2 strongarm card.

1 17. (original) The apparatus of claim 16 wherein the strongarm card is connected to  
2 the Ethernet card via a host CPU circuit.

1 18. (previously presented) The apparatus of claim 14 wherein the third circuit  
2 determines whether the new voice call is to be accepted into the internet protocol  
3 network via the first circuit by comparing said parameter to a plurality of thresholds.

1 19. (previously presented) The apparatus of claim 14 wherein the parameter is a  
2 packet loss ratio defined as

3 
$$PLR = \frac{(\text{lost packets} + \text{late packets})}{(\text{received packets} + \text{lost packets} + \text{late packets})} .$$

1 20. (previously presented) The apparatus of claim 19 wherein the third circuit  
2 compares the packet loss ratio to a lower threshold and if the packet loss ratio is less

3    than the lower threshold, the new voice call is accepted into the internet protocol  
4    network.

1    21. (previously presented) The apparatus of claim 19 wherein the third circuit  
2    compares the packet loss ratio to the lower threshold and an upper threshold, and if  
3    lower threshold < packet loss ratio < upper threshold, the new voice call is accepted  
4    into the internet protocol network at a reduced bandwidth.

1    22. (previously presented) The apparatus of claim 19 wherein the third circuit  
2    compares the packet loss ratio to the upper threshold, and if the packet loss ratio is  
3    greater than the upper threshold, the new voice call is blocked from entering the  
4    internet protocol network.